where α is a parameter, k_0 is a constant integer and P(k) is a weighting window that is symmetrical ~~on both sides of~~ about $k= k_0$.

Claim 2 (Previously Presented): The method according to claim 1, wherein k_0 is equal to zero.

Claim 3 (Currently Amended): The method according to claim 1, further comprising receiving a frequency domain vector (H)~~, and inserting between two coefficients of the vector (H) on each occasion an additional coefficient so as to supply a frequency domain vector (H') of increased length~~.

Claim 4 (Cancelled):

Claim 5 (Cancelled):

Claim 6 (Cancelled):

Claim 7 (Cancelled):

Claim 8 (Currently Amended): The method according to claim ~~7~~ 3, further including computing the coefficients of the filter (H) on the basis of an input signal (X, S1).

Claim 9 (Cancelled)

Claim 10 (Currently Amended): The method according to claim 1, further comprising applying convolution with a first function having the form:

$$U(k) = \sin c\left(\frac{k-k_0}{2}\right) \cdot e^{-j\pi\left(\frac{\alpha(k-k_0)}{2}\right)} \cdot P(k)$$

on a frequency transform (X) of an input signal ~~(S1)~~ that is optionally augmented, and applying a second convolution with a second function having the form:

$$U(k) = \operatorname{sinc}\left(\frac{k-k_0}{2}\right) \cdot e^{-j\pi\left(\frac{\alpha(k-k_0)}{2}\right)} \cdot P(k)$$

on the frequency response (H) of a filter (H) that is optionally augmented, ~~the output vectors (X', S3) from these two modules (M2, M3, M3'') having the same number of coefficients, and in that the method includes multiplying together the coefficients of these two output vectors (X', S3).~~

Claim 11 (Cancelled):

Claim 12 (Previously Presented): The method according to claim 1, wherein the convolution with U outputs a vector B(0,...,N-1) which is such that for all k, a coefficient in B of index k is equal to a product of convolution between Z and U which is such that the coefficient of index k_0 in Z is multiplied in said convolution product with the coefficient of index k_0 of U for which the sinc function has an argument of 0.

Claim 13 (Currently Amended): The method according to claim 1, wherein the function U takes non-zero values over a range of values of k which is symmetrical about the value k_0 ~~for which the modulus of U is at its maximum~~.

Claim 14 (Previously Presented): The method according to claim 1, wherein the function U has an odd number of coefficients Lu, and in that U can be written:

$$U(k) = \operatorname{sinc}\left(-\frac{Lu-1}{4} + \frac{k}{2}\right) \cdot e^{-j\alpha\pi\left(-\frac{Lu-1}{4}+\frac{k}{2}\right)} \cdot P(k)$$

Claim 15 (Currently Amended): The method according to claim 1 <u>further comprising applying convolution with a first function having the form</u>:

$$\underline{U(k) = \operatorname{sinc}\left(\frac{k-k_0}{2}\right) \cdot e^{-j\pi\left(\frac{\alpha(k-k_0)}{2}\right)} \cdot P(k)}$$

on a frequency transform (X) of an input signal that is optionally augmented, and applying a second convolution with a second function having the form:

$$\underline{U(k) = \sin c\left(\frac{k - k_0}{2}\right) \cdot e^{-j\pi\left(\frac{\alpha(k-k_0)}{2}\right)} \cdot P(k)}$$

on the frequency response (H) of a filter (H) that is optionally augmented,~~in combination with claim 10~~, wherein the transform is a discrete Fourier transform.

Claim 16 (Previously Presented): The method according to claim 1, wherein the weighting window is a Kaiser window having a coefficient of 1.5.

Claim 17 (Previously Presented): The method according to claim 1, wherein it constitutes an echo cancelling method.

Claim 18 (Previously Presented): The method according to claim 1, wherein the method constitutes a noise reducing method.

Claim 19 (Previously Presented): The method according to claim 1, wherein α=1.

Claim 20 (Previously Presented): The method according to claim 1, wherein α=-1.

Claim 21 (Currently Amended): An apparatus comprising:

a loudspeaker (100), a microphone (200), an echo canceller (420, 430, 440, 450), and a disturbance reducer (500), the echo canceller including an adaptive filter (470) and a subtracter module (300) delivering the error (Y') between a signal coming from the microphone (200) and a signal obtained by applying the adaptive filter ~~(460)~~ to a loudspeaker signal (100), the adaptive filter (460) adapting its coefficients as a function of said error, and the apparatus including means (495) suitable for transforming the signal from the microphone into the frequency domain upstream from the subtracter module (300) in such a manner that the subtraction is performed in the frequency domain, wherein the apparatus includes means for implementing a method onto the adaptive filter, said method for zeroing a portion of a time domain impulse response of a

filter in a speech for zeroing a portion of a time domain impulse response of a filter in a speech transmission apparatus which filter has a frequency domain transfer function Z(k), said method comprising:

implementing convolution with a function U on the frequency domain transfer function Z(k) where k lies in the range 0 to N-1, wherein the function U has the form:

$$U(k) = \sin c\left(\frac{k-k_0}{2}\right) \cdot e^{-j\pi\left(\frac{\alpha(k-k_0)}{2}\right)} \cdot P(k)$$

where α is a parameter, k_0 is a constant integer and P(k) is a weighting window that is symmetrical ~~on both sides of~~ <u>about k=</u> k_0.

Claim 22 (Previously Presented): The apparatus according to claim 21, further including means (430, 440) for transmitting the result of said frequency domain subtraction to the adaptive filter (470) of the echo canceller.

Claim 23 (Previously Presented): The apparatus according to claim 21, wherein the disturbance reducer (500) is placed downstream from the subtracter module (300) and is applied in the frequency domain to the result of the subtraction.

Claim 24 (Currently Amended): The apparatus according to claim 23, wherein the disturbance reducer (500) includes an adaptive filter (520) suitable for recalculating ~~its~~ coefficients <u>of said disturbance reducer</u> as a function of a frequency domain input signal (Y') from the disturbance reducer (500).

Claim 25 (Previously Presented): The apparatus according to claim 24, wherein the disturbance reducer (500) is placed to receive the frequency domain signal (Y') output from the subtracter module (300) as said frequency domain input signal of the disturbance reducer.

Claim 26 (Previously Presented): The apparatus according to claim 24, wherein the disturbance reducer (500) forms a loop receiving as input the frequency domain signal (Y) output

from the subtracter (300), and applying at its output multiplication by the adapted coefficients of its adaptive filter on the frequency domain signal (Y') output by the subtracter (300).

Claim 27 (Previously Presented): The apparatus according to claim 24, wherein the same frequency domain signal (Y') is used as an error signal for adapting the adaptive filter (470) of the echo canceller and is multiplied by the coefficients of the adaptive filter (520) of the disturbance reducer (500).

Claim 28 (Currently Amended): The apparatus according to claim 21, ~~whereiin~~ wherein no transform module is placed between the subtracter module (300) and the disturbance reducer (500).

Claim 29 (Previously Presented): The method according to claim 1, wherein the filter is an adaptive filter of an echo canceller in an apparatus comprising a microphone and a loudspeaker.

Claim 30 (Currently Amended): A method for zeroing a time domain speech transmission signal which signal has a frequency domain transform Z(k), said method comprising implementing convolution with a function U on the frequency domain transform Z(k) where k lies in the range 0 to N – 1, wherein the function U has the form:

$$U(k) = \sin c\left(\frac{k - ko}{2}\right) \cdot e^{-j\pi\left(\frac{\alpha(k-ko)}{2}\right)} \cdot P(k)$$

where α is a parameter, K_0 is a constant integer and P(k) is a weighting window that is symmetrical ~~on both sides of~~ about k=ko K_0.

Claim 31 (Currently Amended): The method according to claim 30, further including a transform into the frequency domain of an input time domain signal (S1), upstream from performing convolution with U, ~~and in that Z is said frequency transform (X), possibly associated with receiving a frequency domain vector (H), and inserting between two coefficients of the vector (H) on each occasion an additional coefficient so as to supply a frequency domain vector (H') of increased length~~.